IN THE UNITED STATES PATENT OFFICE

PATENT APPLICATION

ECHO/NOISE CANCELING DEVICE FOR USE WITH PERSONAL COMPUTERS

This device relates to the computer industry and specifically to a device to reduce or cancel echo and noise found within internet telephony.

This application claims priority from Provisional Patent Application Serial Number 60/197,555 filed on April 17, 2000.

BACKGROUND OF THE INVENTION

The speakerphone device has been in use for years on traditional telephone lines around the world. This type of device generally is an instrument utilizing a switching technology that allows one or the other party to speak at once, but not simultaneously. There are more advanced devices that integrate DSP's (Digital Signal Processors) to accomplish a Full-Duplex (simultaneous conversation), but they are very expensive by comparison. See for example U.S. Patent 5,657,384, Staudacher et al., which discloses a Full Duplex Speakerphone using DSP technology.

The invention presented in this disclosure is an inexpensive Full-Duplex solution for the new and exciting Internet using standard analog techniques. Numerous

companies are offering free long-distance phone calls over the Internet, but users are mostly restricted to using headphones, due to echo. DSP technology is prohibitive, due to the long delay before the echo occurs. This long delay would require a rather large amount of memory and CPU overhead resulting in too expensive of a device for the consumer market.

The invention presented in this disclosure accomplishes a Full-Duplex conversation, without echo, over the Internet at a very inexpensive price. The invention makes an Internet conversion duplicate the natural sound of a face-to-face conversion.

SUMMARY OF THE INVENTION

There is described in the present disclosure a speakerphone adapter for suppressing general household background noise, i.e. air conditioners, radios and televisions. This device also significantly reduces echo and regenerative feedback associated with microphone and speaker devices. The device also increases the effective digital communications bandwidth over the Internet by improving on the silence detection compression algorithms through the use of hardware. The invention is positioned between the microphone and speakers and the personal computer. The device in the present disclosure is comprised of a microphone pre-filter, amplifier, band-pass filter, a microphone attenuator and a post filter; a mute control amplifier, mute control filter, an absolute value circuit, a successive peak integrator, a R/C control circuit and a mute control comparator. The device also is comprised of a Right channel attenuator, Left channel attenuator, a receive signal combiner, signal filter amplifier, an absolute

value circuit and a successive peak integrator amplifier.

The pre-filter, microphone amplifier and band-pass filter provide the correct shape and amplitude of the transmitted audio signal. The shape can be represented by a filter response having a center frequency of 638 hertz with –3dB cutoff points at 281 hertz and 1.5 kilohertz. The microphone attenuator provides up to 9.8dB of signal attenuation. The post-filter smoothes the overall audio signal after attenuation and muting occurs. The device also consists of a mute control amplifier and filter providing gain and reduction of unwanted signals. The filter response can be represented with a pass-band center frequency of 1.17 kilohertz and –3dB points at 447.9 hertz and 2.9 kilohertz. The filtered signal is then amplified and integrated over time driving a R/C timing circuit. Finally the R/C timing circuit is the signal source for the mute control comparator, which is used for improving the digital signal bandwidth.

The Left and Right channel attenuators, receive signal combiner, the signal filter amplifier, absolute value circuit and integrator amplifier form the receive side of the device. The channel attenuators provide up to 9.8dB of attenuation individually. The receive signal combiner is utilized for combining the Left and Right channels to feed the signal filter and amplifier. The filter portion can be represented by a filter response having a high-pass frequency of 1.0 kilohertz with a –3dB cutoff at 175 hertz. The gain of the amplifier is approximately 20dB. The amplified and filtered signal is then feed into the ABS (absolute value circuit). The varying DC signal developed in the ABS circuit is then amplified and successively integrated over time providing the microphone attenuator control signal.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 shows a block diagram of the transmit section of the instant device.

Figure 2 shows a block diagram of the microphone mute section of the instant device.

Figure 3 shows a block diagram of the receive section of the instant device.

Figure 4 is page one of the general device schematic diagram.

Figure 5 is page two of the general device schematic diagram.

Figure 6 shows the microphone pre-filter response curve.

Figure 7 shows the second order band-pass filter response curve.

Figure 8 shows the post filter response curve.

Figure 9 shows the overall microphone circuit response curve.

Figure 10 shows the mute control amplifier second order band-pass filter response curve.

Figure 11 shows the mute control filter response curve.

Figure 12 shows the overall mute control response curve.

Figure 13 shows the R/C Control Decay Curve.

Figure 14 shows the Signal Filter Amplifier response curve.

DESCRIPTION OF THE INVENTION

Figure 1 is a block diagram illustrating the Transmit Section of the device. Beginning with the Microphone Input Port the microphone signal 100 is applied to Pre-Filter, **1A**. (The microphone input jack is shown in Figure 4, and only one channel is used. It should be noted that both left and right channels may be combined through a proper resistive network.) The Pre-Filter, **1A**, has a band-pass filter response with the preferred center frequency of 2.3 kilohertz with –3dB points at 287 hertz and 19.28 kilohertz, although the values may vary by plus/minus 20 percent. The filter response curve can be seen in Figure 6. The signal then continues on path 110 to Microphone Amplifier, **1B**.

The Microphone Amplifier, **1B**, has an AC signal gain of approximately 32.6 dB producing signal 120. The output signal, 120, enters a Second Order Band-pass Filter, **1C**, having a preferred center frequency of 638 hertz with –3dB cutoff points at 281 hertz and 1.5 kilohertz, although the values may vary by plus/minus 20 percent. The filter response curve can be seen in Figure 7.

The filtered signal, 130, continues onto the Resistive Isolator, **1D**, which is a simple resistor designed to isolate signal 130 and signal 140 from the Microphone Attenuator circuit, **1E**. Signal 140 also provides the **AUDIO-MIX** signal that serves as the input signal to the Microphone Mute Control Circuit shown in Figure 2.

The Microphone Attenuator circuit, **1E**, provides a maximum attenuation of approximately 9.8dB. The attenuator circuit employs a bipolar transistor, **Q1**, which is connected by its base to the microphone attenuator control signal, **MIC-ATT**, shown as

signal 360. Signal 360 is one of the outputs of the Receive Circuit that will be described later in the description of Figure 3. Essentially, signal 360 controls the Collector-Emitter resistance of **Q1**, which in turn operates with resistors **R4** and **R7**, as a voltage divider, producing attenuated signal 140 (see Figure 4), which passes through the Capacitive Isolator circuit, **1F**, to the Post-Filter circuit, **1G**.

The capacitive isolator circuit is a simple capacitor, **C3**, which serves to isolate signal 140 from signal 160. Signal 160 can effectively be shunted to ground by the microphone mute control signal, **XMUTE**, (signal 261) which is one of the outputs of the Microphone Mute Control circuit shown in Figure 2.

The Post-Filter, **1G**, is a high-pass filter having a preferred frequency response of 1.45 kilohertz and greater pass-band with a –3dB cutoff point at 722 hertz, although the values may vary by plus/minus 20 percent. The filter response curve can be seen in Figure 8. Following the high-pass filter is the Microphone Output Port. (Both channels of the output jack may be connected together as shown in Figure 4.) The overall frequency characteristics of the combined filter response's within the Microphone Attenuator circuit can be seen in Figure 9.

Figure 2 is an illustrated block diagram of the Microphone Mute Control Section. The input signal is derived from the resistive isolator of Figure 1 as the **AUDIO-MIX** signal, signal 140 (Figure 1) that is also signal 200 (Figure 2), and is applied to the Mute Control Amplifier, **2A**. The Mute Control Amplifier, **2A**, exhibits an AC signal gain of approximately 39dB. The amplifier also has a band-pass filter response having a preferred center frequency of approximately 4.2 kilohertz with –3dB cutoff points at 1.4 kilohertz and 12.5 kilohertz, although the values may vary by plus/minus 20 percent, that

produces signal 210. The filter response curve can be seen in Figure 10. Signal 210 is applied to the Mute Control Filter, **2B**.

The Mute Control Filter, **2B**, is a Second Order Low-Pass Filter having a preferred frequency of 723 hertz with a –3dB cutoff point at 1.25 kilohertz, although the values may vary by plus/minus 20 percent. The slope of the filter can be seen in Figure 11. More important is the combined overall frequency response of the Mute Control Amplifier, **2A**, and the Mute Control Filter, **2B**. The preferred combined band-pass filter response, signal 220, has a center frequency of 1.17 kilohertz with –3dB cutoff points at 447.9 hertz and 2.9 kilohertz, although the values may vary by plus/minus 20 percent. The preferred filter response curve can be seen in Figure 12.

The band-pass filtered signal 220 passes onto the ABS (Absolute Value) circuit, **2C**. The ABS circuit, **2C**, can be recognized as a ladder-type single stage absolute-value circuit as described by H. A. Wittlinger in the RCA CA3140 data sheet ("Applications"). The gain of the ABS circuit is 0.5, thus a 2 Volt peak-to-peak signal would produce an absolute value of 1 Volt. The rectified DC signal 230 enters an Integrator Amplifier circuit, **2D**.

The Integrator Amplifier, **2D**, has an RC charge time constant of approximately 29 milliseconds and a RC discharge time constant of approximately 941 milliseconds, although the values may vary by plus/minus 20 percent. The amplifier portion of the Integrator Amplifier, **2D**, has a DC signal gain of approximately 21dB, although the value may vary by plus/minus 20 percent. The varying DC signal 240 produced continues onto R/C Control Circuit, **2E**.

The R/C Control circuit, **2E**, is used to extend the discharge time of Integrator

Amplifier, **2D**, and smooth the decay curve. A typical decay curve time would be 6.5 seconds and an example of the decay curve can be seen in Figure 13. The output of the R/C Control circuit, signal 250, forms the speaker attenuator control signal, **SPK-ATT**, and also drives the Mute Control Circuit, **2F**. The **SPK-ATT** signal controls the output of the speakers and essentially attenuates the speakers whenever audio is present at the microphone.

The Mute Control circuit, 2F, is a voltage comparator with a trip voltage point of approximately 1.0 volt. Once the trip point is exceeded the comparator enables the microphone mute control signal, XMUTE (signal 261). XMUTE (as explained above) allows or disallows the microphone audio signal developed in the Microphone Attenuator circuit, 1E, signal 160, to pass onto the Microphone Output (signals 170-171). The time and voltage relationship of the comparator is shown in Figure 13 as the "Mute Trip Point" that occurs when the voltage reaches 1.0 VDC at 1.5 seconds in time.

The **XMUTE** signal suppresses all transmitted audio from the device to the PC. This is because some software used within personal computers for CHAT and IP Telephony directly convert any audio signal. I.e., small amounts of audio received at the Microphone Input Port that are useless for all intents and purposes will be heard or "seen" by the software. The problem is that, once the CODEC begins converting useless sound into useless data, the useless data starts utilizing bandwidth thus degrading the quality of the conversation given finite bandwidth restrictions.

Figure 3 is an illustrated block diagram of the Receive Section of the device. Beginning with the Left Channel In Port the audio signal, 300, enters the Left Channel Attenuator, **3AL**. The attenuated signal, 302, exits Left Channel Attenuator via the Left

Channel Output Port. Now examining the Right Channel In Port the audio signal, 301, enters the Right Channel Attenuator, **3AR**. The attenuated signal, 303, exits Right Channel Attenuator via the Right Channel Output Port. The attenuation of both channels is controlled by the **SPK-ATT** signal, 250, provided by the Microphone Mute Control Section described above. The **SPK-ATT** signal controls the output of the speakers and essentially attenuates the speakers whenever audio is present at the microphone (the user is speaking into the microphone) and provides a maximum attenuation of approximately 9.8dB.

The un-attenuated Left Channel Signal 300/320 passes onto the Receive Signal Combiner, **3B**, along with the un-attenuated Right Channel Signal 301/321. The Receive Signal Combiner, **3B**, is a resistive network mixing the Left Channel with the Right Channel using resistors **R13** and **R17**. (See Figure 4.) The combined signal. 330, exits the Signal Combiner, **3B**, and travels to the Signal Filter Amplifier, **3C**.

The Signal Filter amplifier, **3C**, has a preferred high-pass filter response of approximately 1 kilohertz with a fairly steep low-pass cutoff of –3dB at 175 hertz as can be seen in Figure 14, although the values may vary by plus/minus 20 percent. The AC signal gain is approximately 20dB. The filtered signal continues on as signal 340 entering the ABS (Absolute Value) circuit, **3D**. The ABS circuit can be recognized as a ladder-type single stage absolute-value circuit described by H. A. Wittlinger in the RCA CA3140 data sheet ("Applications"). The gain of the ABS circuit is 0.5, thus a 2 Volt peak-to-peak signal would produce an absolute value of 1 Volt. The rectified DC signal, 350, drives the Integrator Amplifier, **3E**.

The Integrator Amplifier, **3E**, has an RC charge time constant of approximately

4.7 milliseconds and a RC discharge time constant of approximately 75 milliseconds, although the values may vary by plus/minus 20 percent. The amplifier portion of the Integrator Amplifier, **3E**, has a DC signal gain of approximately 20dB. The varying DC signal produced by the Integrator Amplifier forms signal 360, the microphone attenuator control signal, **MIC-ATT**, that controls the Microphone Attenuator, **1E**, of the Transmit Section (Figure 1), as previously described. Essentially, whenever there is audio in the speaker, the microphone audio output signal is attenuated.

Signal 330 applied to the Signal Filter Amplifier, **3C**, the ABS Circuit, **3D**, the Integrator Amplifier, **3E**, essentially form the echo logic circuit that controls attenuation of the microphone in the transmit section of the device.

As previously explained, whenever there is audio in the microphone, the speakers are attenuated, and whenever there is audio in the speaker, the microphone audio output signal is attenuated. There is, however, interaction between speaker attenuation, microphone attenuation and microphone muting, but the combined analog delays found in the filter responses and within the three sections forming the overall invention act together to cancel echo.

Figures 4 and 5 show the overall device schematic diagram and the signals referenced in the above description are shown in these schematics. The average person skilled in the art should be able to take these schematics, which are the actual schematics used in the working prototype device and duplicate the device, providing reference is made to the circuit description given above.

The power supply, not described in detail, is shown in these schematics, and any person skilled in the art may readily duplicate the power supply portion of the preferred

device. The power supply takes DC power (9VDC) supplied by a commercially available mains to dc unit, regulates the supplied DC, and supplies the required power to components within the device.

The device is packaged in a small enclosure with standard jack connections for interface with soundcards found on personal computer systems (or the like). In some circumstances power may be taken from the computer system; otherwise, power may be supplied from the AC line (AC Mains) using standard techniques described above. Other forms of jacks, connections and/or enclosures may be used to interface the device to other systems. In fact the device may be built-in to systems which require the characteristics exhibited and described in this disclosure. The discrete component circuit of the preferred device, may easily be implemented in an large scale integrated circuit or in a hybrid form. This is a manufacturing choice.

There is disclosed the best and preferred mode for practicing the art described in this disclosure. It should be realized that a person skilled in the art may change the individual filter response curves to obtain the same general overall response curves. As stated in the disclosure it is the overall response curves that cause the instant invention to operate in the manner described. Therefore, any such changes would be contemplated by the inventor and fall within the claims and the scope of the disclosure.